FORM PTO-1390 (Modified)
(REV 11-2000)

U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE

TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371

ATTORNEY'S DOCKET NUMBER
112740-388

531 Rec'd PCT/PTO 14 JAN 2002

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR
10/031292

INTERNATIONAL APPLICATION NO.	INTERNATIONAL FILING DATE	PRIORITY DATE CLAIMED
PCT/DE00/02303	14 July 2000	14 July 1999

TITLE OF INVENTION
METHOD AND APPARATUS FOR DECODING SOURCE SIGNALS

APPLICANT(S) FOR DO/EO/US
Stefan Heinen et al.

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)). The submission must include itens (5), (6), (9) and (24) indicated below.
4. ☒ The US has been elected by the expiration of 19 months from the priority date (Article 31).
5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))
   - a. ☒ is attached hereto (required only if not communicated by the International Bureau).
   - b. ☐ has been communicated by the International Bureau.
   - c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☒ An English language translation of the International Application as filed (35 U.S.C. 371(c)(2)).
   - a. ☒ is attached hereto.
   - b. ☐ has been previously submitted under 35 U.S.C. 154(d)(4).
7. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))
   - a. ☐ are attached hereto (required only if not communicated by the International Bureau).
   - b. ☐ have been communicated by the International Bureau.
   - c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   - d. ☒ have not been made and will not be made.
8. ☐ An English language translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
9. ☒ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).
10. ☐ An English language translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).
11. ☒ A copy of the International Preliminary Examination Report (PCT/IPEA/409).
12. ☒ A copy of the International Search Report (PCT/ISA/210).

Items 13 to 20 below concern document(s) or information included:

13. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
14. ☒ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
15. ☒ A **FIRST** preliminary amendment.
16. ☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.
17. ☒ A substitute specification.
18. ☐ A change of power of attorney and/or address letter.
19. ☐ A computer-readable form of the sequence listing in accordance with PCT Rule 13ter.2 and 35 U.S.C. 1.821 - 1.825.
20. ☐ A second copy of the published international application under 35 U.S.C. 154(d)(4).
21. ☐ A second copy of the English language translation of the international application under 35 U.S.C. 154(d)(4).
22. ☒ Certificate of Mailing by Express Mail
23. ☐ Other items or information:

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PCTUS1/REV03

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR	INTERNATIONAL APPLICATION NO.	ATTORNEY'S DOCKET NUMBER
10/031292	PCT/DE00/02303	112740-388

24. The following fees are submitted:.

CALCULATIONS PTO USE ONLY

531 Rec'd PCT/P 14 JAN 2002

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2)) paid to USPTO and International Search Report not prepared by the EPO or JPO **$1040.00**

☒ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but International Search Report prepared by the EPO or JPO **$890.00**

☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO **$740.00**

☐ International preliminary examination fee (37 CFR 1.482) paid to USPTO but all claims did not satisfy provisions of PCT Article 33(1)-(4) **$710.00**

☐ International preliminary examination fee (37 CFR 1.482) paid to USPTO and all claims satisfied provisions of PCT Article 33(1)-(4) **$100.00**

ENTER APPROPRIATE BASIC FEE AMOUNT =				**$890.00**
Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)).				**$0.00**
CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE	
Total claims	15 - 20 =	0	x $18.00	**$0.00**
Independent claims	2 - 3 =	0	x $84.00	**$0.00**
Multiple Dependent Claims **(check if applicable).**			☐	**$0.00**
TOTAL OF ABOVE CALCULATIONS =				**$890.00**
☐ Applicant claims small entity status. See 37 CFR 1.27). The fees indicated above are reduced by 1/2.				**$0.00**
SUBTOTAL =				**$890.00**
Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)). +				**$0.00**
TOTAL NATIONAL FEE =				**$890.00**
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable).** ☐				**$0.00**
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a. ☒ A check in the amount of **$890.00** to cover the above fees is enclosed.

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NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

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REGISTRATION NUMBER

January 14, 2002

DATE

BOX PCT

IN THE UNITED STATES ELECTED/DESIGNATED OFFICE
OF THE UNITED STATES PATENT AND TRADEMARK OFFICE
UNDER THE PATENT COOPERATION TREATY-CHAPTER II

PRELIMINARY AMENDMENT

APPLICANTS:	Stefan Heinen et al.	DOCKET NO.:	112740-388
SERIAL NO:		GROUP ART UNIT:	
FILED:		EXAMINER:	
INTERNATIONAL APPLICATION NO::		PCT/DE00/02303	
INTERNATIONAL FILING DATE		14 July 2000	
INVENTION:	METHOD AND APPARATUS FOR DECODING SOURCE SIGNALS		

Assistant Commissioner for Patents,
Washington, D.C. 20231

Sir:

Please amend the above-identified International Application before entry into the National stage before the U.S. Patent and Trademark Office under 35 U.S.C. §371 as follows:

In the Specification:

Please replace the Specification of the present application, including the Abstract, with the following Substitute Specification:

SPECIFICATION

TITLE OF THE INVENTION

METHOD AND APPARATUS FOR DECODING SOURCE SIGNALS

BACKGROUND OF THE INVENTION

The present invention relates to a method for decoding source signals, which have been transmitted in coded form via a transmission channel. The present invention furthermore relates to a corresponding apparatus for decoding the source signals. The expression source signals refers to, for example, voice, audio or video signals. The following text is largely based on voice signals. In this case, however, voice signals should be regarded only as an example and, in this respect, are not associated with any restriction whatsoever to the present invention.

Methods and apparatuses such as these are currently used, typically, for voice transmission in mobile radio networks. Mobile radio networks are generally set up such that a number of receivers or receiver/transmitter stations, which are referred to as base transceiver stations (and also as BTS in the following text) are arranged as far as possible covering an area at predetermined intervals from one another, so that the reception areas of these BTS overlap. A number of these BTS are normally linked via cables to a common base station controller (also referred to as a BSC in the following text), with the distance between the BTS and the BSC generally being several kilometers. In turn, these BSC are then, generally, connected to a mobile switching center (also referred to as an MSC in the following text), which is responsible, inter alia, for call management, call monitoring and for interaction with other networks; for example, normal landline networks or the ISDN.

For transmission via a radio path, the voice signals are initially broken down in a source coder, in this case a voice coder, in time voice sections, which can be described by different speech parameters. These real-value speech parameters are then quantized. A parameter quantized in this way corresponds to an entry in a quantization table or a code book, with the parameter being represented by a specific bit combination. The speech parameters which have been digitally coded in this way are then passed on. Other types of source signals are broken down in a similar way into source-coded parameters or else source-coded coefficients, which are then passed on.

Since interference and losses must be expected when transmitting the parameters via a mobile radio channel, redundant information is added to these coded signals in a channel coder, from which information it is possible to draw conclusions about the correctness of the received signal, at the receiver end.

The data received via the radio channel is then initially decoded in a corresponding channel decoder, with all the data which was required only for transmission on the radio channel being separated out again. This includes, inter alia, the redundant information, which contains details about the transmission quality. This channel decoder is generally located at the same place as the BTS. The speech parameters are, finally, passed on for final decoding to a source decoder (or voice decoder). This source decoder is normally a component of a TRAU (Transcoder/Rate Adapter Unit), which is arranged physically separately from the BTS, typically in the BSC or MSC. As such, the data rate between the BTS and the BSC or the MSC is kept low, thus making it possible, inter alia, to reduce the transmission costs.

When transmitting voice data via a radio channel that is subject to severe interference, residual bit errors can still remain in the bit stream despite error correction in the course of the channel decoding process. Voice decoding based on these speech parameters that are subject to interference would generally lead to considerable distortion of the output voice signal. Additional error concealment measures are therefore required, in order to improve the subjectively perceived reproduction quality, with the information determined by the channel decoder about the respective transmission quality or the reliability of the decoded bits sensibly being used for this purpose.

In the already existing GSM Standard, channel-dependent reliability information, referred to as a BFI flag (bad frame indication), is produced in the channel decoding process for each voice frame; that is, for each received bit sequence. To do this, the channel decoder carries out a CRC (Cyclic Redundancy Check), which is formed by the auditively most important bits of the speech parameter indices or those bits which are most sensitive to interference. This check results in a binary decision (BFI=0, good frame; BFI=1, bad frame), which requires only one bit. This flag is passed on to the voice decoder, where it is included in the decoding process in such a way that, for a frame which is identified as being bad, the values of the last good frame are repeated directly or in slightly modified form; for example, by being attenuated. Muting is carried out in the event of a number of successive bad frames as a result of which, in the event of severe channel interference, so many signal components are cut off in some circumstances that the comprehensibility is considerably adversely affected.

T. Fingscheidt and P. Vary have proposed a completely new error concealment method in the article "Error Concealment by Softbit Speech Decoding", in the Proceedings of the ITG Conference Voice Communication, pages 7-10, Frankfurt am Main, September 1996. In this method, the channel-dependent reliability information determined by the channel decoder is used in order to calculate the probability with which a specific speech parameter has been transmitted, or in order to establish an estimated value for that speech parameter such that it matches the actually transmitted speech parameter as well as possible. In detail, a calculation is, in this case, carried out for each potentially transmitted bit combination to determine the probability with which this bit combination can be changed to the received bit combination. The reliability information is, in this case, determined in the form of individual bit error probabilities, with one, and only one bit error probability being associated with each individual information bit. The pair, including the information bit itself and the associated bit error probability, is referred to as a softbit. These softbits must be

made available to the error concealment device in order to estimate the speech parameter. In this type of error concealment, the estimation results in different real-value parameter values (intermediate values) than those contained in the transmission-end quantization table, so that the estimated parameter value cannot be coded correctly using this quantization table. For this reason, this device normally has, until now, been connected directly to the voice decoder, since it can process the estimated real-value parameter values directly without any additional losses.

However, this leads to difficulties at the base station end. For the reasons mentioned above, the channel decoder, which obtains the reliability information from the transmitted signals, is located at the same point as the receiver station (BTS) while the voice decoder is located at the TRAU, with the transmission rate being limited on the connection between them. With the current GSM Standard, the limit is 16 or 8 kbit/s (full rate or half rate), which corresponds to 16 or 8-sub-multiplexing, respectively. However, if a softbit is represented, for example, by 4 bits in an error concealment process based on the last described method, this would result in a data stream at a total of 12.2 x 4 = 48.8 kbps when using the normal 12.2 kbps voice coders and decoders in accordance with the GSM Standard. Transmission via a data line which is limited to 16 kbps is thus impossible.

An object of the present invention is, therefore, to provide an alternative to this prior art.

SUMMARY OF THE INVENTION

A key feature of the inventive method and apparatus is the decoupling of the error concealment from the actual source decoding. By carrying out the error concealment at the same location as the channel decoding, or the receiver, there is no longer any need to pass on additional channel status information to the source decoder in addition to the source-coded parameters, so that the data stream can be reduced considerably. It is thus possible, even with the existing systems and with the previous standards, to carry out an improved error concealment method, which uses the channel-dependent reliability information more effectively.

The source-coded parameters which are estimated in the error concealment method can, in principle, be passed on to the source decoder in any desired manner. However, preferably, these parameters are quantized in the same way as the transmitted source-coded parameters and are passed on in the known digital form to the source decoder in order to minimize the transmitted data streams. If the quantization is chosen in an appropriate suitable manner, it has been found that this renewed quantization, referred to as

requantization in the following text, does not result in any additional losses or interference whatsoever.

In particular, it has been found that it is advantageous if at least the same quantization steps as those used for quantization of the originally transmitted source-coded parameters are used for quantization of the estimated source-coded parameters. In the event of error-free transmission, the transmitted source-coded parameters are automatically reproduced exactly. If this were not the case, then errors could occur due to the requantization in the event of transmission without interference.

Various investigations and simulations have shown that, when using quantizers with a sufficiently high quantization rate R, it is sufficient to use exactly the same quantizers as those at the transmission end for requantization of the estimated parameters. If the parameters have a Gaussian distribution, this is generally possible for $R > 1$ bit/sample value for transmitter-end quantizers. The Gaussian distribution criterion is, in this case, to an approximation, quite well satisfied by most of the transmitted source-coded parameters. The code book which is used at the transmission end thus can be used as the quantization code book for the estimated parameter.

If a quantizer with a rate of $R \leq 1$ bit/sample value is used at the transmission end, any influence from the quantization of the estimated parameters can be ameliorated by using a modified quantizer at a higher rate at this point. As such, a quantizer is chosen which uses further quantization steps in addition to the quantization steps which are available at the transmission end. For example, if a quantizer with a rate of only $R = 1$ bit/sample value is used at the transmission end, it is worthwhile using a quantizer with a rate of $R = 2$ bits/sample value for quantization of the estimated parameters. This allows additional interference due to requantization to be avoided very well at the receiver end, even in cases such as this, in a simple manner.

In the course of the further development of the GSM mobile radio standard, a new standard for coded voice transmission has now been developed. This relates to coders and decoders (codecs) which allow the overall available data rate to be split in a manner matched to the channel state and to the system load (adaptive multirate codecs; AMR codecs). In this case, the channel mode is defined on the one hand (full rate 22.8 kbps or half rate 11.4 kbps) while, on the other hand, the source and channel coding are matched to the data rate that is then available. As such, the coding rate is varied during a transmission as a function of the transmission quality of the channel and the number of users who are using this channel at the

same time. The number of quantization steps for the coding process is also changed in a corresponding manner. It is, thus, advantageous for the quantization of the source-coded parameters, which are estimated using the error concealment method, to be carried out as a function of the present coding rate of the transmitter as well. This is preferably done using a quantizer which is matched to the various coding rates of the transmitter.

Additional features and advantages of the present invention are described in, and will be apparent from, the following Detailed Description of the Invention and the Figures.

BRIEF DESCRIPTION OF THE FIGURES

Figure 1 shows a schematic illustration of the position and connections for the receivers (BTS), the BSC and the mobile switching station (MSC).

Figure 2 shows a schematic illustration of the signal path through the individual coders and decoders.

DETAILED DESCRIPTION OF THE INVENTION

As is shown in simplified form in Figure 1, a conventional modern base station operating in accordance with the GSM mobile radio standard includes a BSC 11, to which a number of BTS 6 are connected via data lines 14. The BSC 11 is, in turn, connected via a data line 15 to an MSC 13. In general, one MSC 13 serves a number of BSC 11, although this is not illustrated in Figure 1, for space reasons.

The BTS 6 are located spaced apart from one another such that their reception areas 17 just overlap, so that the supply from the BTS 6 covers the area as well as possible. If a mobile telephone 1 is now located in the reception area of one BTS 6, then it can communicate with this BTS 6 via a radio path 17. When an active mobile telephone 1 leaves the reception area 17 of one BTS 6 and enters the reception area 17 of another BTS 6 associated with the same base station, this is identified automatically by the associated BTS 11, and a handover takes place from the one BTS 6 to the next. During a handover from the area of one base station to another base station, the handover process is carried out with the aid of the MSC 13. As such, the call is automatically handed over to the new base station, with the other BSC 11.

The details of the profile of a voice signal from a mobile telephone 1 to the BSC 11, in particular the different coding and decoding stages, can be seen from the simplified schematic illustration in Figure 2. This shows only the signal interchange in one direction. Normally, the coders and decoders are each codecs, that is to say combined coding/decoding devices, and the transmitting and receiving units are combined transmitting/receiving units (transceivers).

As illustrated in Figure 2, the voice signal is first of all passed through a voice coder 2, in which the speech is broken down into individual speech parameters. Each sound is, in this case, represented by a specific number of speech parameters. Typical speech parameters in one representation of the voice signals are, for example, the "LPC coefficient," the "LTP index," the "LTP gain," the "codebook indices" and the "codebook gain," and in another representation are the "LSP set," the "pitch delay," the "pitch gain," the "algebraic code" and the "codebook gain."

These speech parameters are then passed through a quantizer 3, where they are converted to a bit combination; that is to say, a real-value speech parameter v is represented by the bit combination X after quantization. Depending on the transmission rate, a different number of bits are available for coding for the individual parameters. In the AMR Standard, eight different transmission modes, with data rates of between 12.2 kbps and 4.75 kbps can be used for voice transmission.

The speech parameters v are transmitted from the quantizer 3, in the digital representation X, to a channel coder 4, which adds the channel information, required for transmission, to the data. This includes redundant data, which allows the receiver to check the correctness of the received data and, if necessary, to correct transmission errors. The radio signals are then sent from a transmission unit 5 or a transceiver via a radio channel 16 to a receiving element 7, for example an antenna with a demodulator and/or equalizer, of the BTS 6.

In the BTS 6, the received signals S are first of all passed to a channel decoder 8, which initially decodes the received signals S. During the initial decoding process, reliability information which is correlated with the transmission quality is obtained from the signals S. This may include, for example, the results of parity checks.

This additionally obtained information about the channel state is transmitted together with the received speech parameters determined in the channel decoder 8 to an error concealment device 9. This is indicated in Figure 2 by two arrows between the channel decoder 8 and the error concealment device 9. The speech parameter received via the radio channel is still available in digitally coded form at the output of the channel decoder 8, that is to say it is represented by a bit combination X', which matches the transmitted bit combination X, provided the transmission has taken place without errors.

In the present exemplary embodiment, the error concealment device 9 operates in accordance with the error concealment method cited in the prior art by T. Fingscheidt and P. Vary. In this method, the reliability information is initially used to determine, for each

individual bit in the combination X, the probability that it has been transmitted without any errors. This probability is dependent solely on the state of the channel. In addition, the error concealment method can also make use of information which is dependent on the original source of the received signal. This may be, for example, the probabilities with which a specific parameter, and hence a specific bit combination, can occur at the transmission end. These probabilities of occurrence can be determined in advance via a representative speech database, and can be stored in tables. Furthermore, it also would be possible to take account of the probability of two specific parameters or bit combinations following one another directly, in order to take account of correlations between successive received voice frames.

The individual probabilities, which are either channel-dependent or source-dependent, are then used either to estimate a parameter $\hat{v}$, which most probably corresponds to the originally transmitted speech parameter v (maximum a-posteriori estimation) or an estimate is made, in which the mean square error of the possible errors between the estimated parameter $\hat{v}$ and the transmitted parameter v is a minimum (mean-square estimation).

The real-value parameter $\hat{v}$ estimated by the error concealment device 9 is then once again passed (before being transmitted to the voice decoder 12 arranged at the BSC 11) through a quantizer 10 which operates in the same way as the quantizer 3 at the transmission end and converts the real-value parameter $\hat{v}$ to a digital bit combination $\hat{X}$. Instead of the bit combination X' determined by the channel decoder for the received parameter, the voice decoder 12, in consequence, now receives a bit combination $\hat{X}$, which is coded in the same way and represents the speech parameter $\hat{v}$ which is estimated by the error concealment and which most probably matches the transmitted speech parameter v, which differs from it by the smallest error.

The quantizer 10 in the BTS in the present exemplary embodiment is precisely the same model as the quantizer 3 in the mobile telephone 1. These are normal codecs; for example, AMR or FR (full rate) codecs to the GSM Standard. The codebooks available to the quantizer 10 are also the same as those available to the quantizer 3.

It should be mentioned once again that the present invention is not restricted to the specific exemplary embodiment described. Thus, in principle, it is also possible to use it in systems which do not operate in accordance with the GSM Standard, but, for example, in accordance with more recent Standards that are currently still being developed, such as the UMTS Standard. In the same way, the source decoder 12 need not necessarily be located in the BSC 11, but can be arranged as an autonomous unit, on its own or combined with other

functional units, for example in the form of what is referred to as the TRAU, upstream or downstream of the BSC. In particular, instead of the method according to Fingscheidt and Vary, it is also possible to use a different error concealment method, which uses the reliability information to estimate the transmitted source-coded parameter, that is to say which corrects the received parameter using the reliability information, such that it corresponds to the supposedly correct transmitted parameter. Furthermore, as already mentioned above, the present invention is not restricted to voice signals, but also can be used for any other desired source signals.

The method and the apparatus according to the present invention result in improved transmission quality even in very poor radio channels, since the channel-dependent reliability information received by the channel decoder and the source-dependent information are used in a considerably better way to eliminate errors that occur. Owing to the specific physical arrangement of the error concealment device and the downstream quantizers, this can be done without increasing the data rate between the channel decoder and the source decoder.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

ABSTRACT OF THE DISCLOSURE

A method and an apparatus for decoding coded source signals which are transmitted via a transmission channel. In this case, the received signals are initially decoded in a channel decoder and, in the process, source-coded parameters are determined from the received signals and are passed on to a source decoder, which is physically separated from the channel decoder, where they are further-decoded. Reliability information, which is correlated with the transmission quality, is obtained from the received signals during the initial decoding. The transmitted source-coded parameters are estimated from the received source-coded parameters via this reliability information using an error concealment method. The error concealment is carried out at the same location as the channel decoder, and the estimated source-coded parameters are passed on to the source decoder.

In the claims:

On page 13, cancel line 1, and substitute the following left-hand justified heading therefor:

CLAIMS

Please cancel 1-15, without prejudice, and substitute the following claims therefor:

16. A method for decoding source signals, the method comprising the steps of:

decoding received signals initially in a channel decoder;

obtaining reliability information, which is correlated with transmission quality, from the received signals during the initial decoding;

determining source-coded parameters from the received signals;

passing on the source-coded parameters to a source decoder which is physically separated from the channel decoder; and

further decoding the source-coded parameters at the source decoder;

wherein the transmitted source-coded parameters are estimated from the received source-coded parameters via the reliability information using an error concealment method, the error concealment being carried out at a same location as the channel decoder, and the estimated source-coded parameters are passed on to the source decoder.

17. A method for decoding source signals as claimed in claim 16, wherein the estimated source-coded parameters are quantized for passing on to the source decoder.

18. A method for decoding source signals as claimed in claim 17, wherein quantization steps used for quantization of the transmitted source-coded parameters are the same as those used for the quantization of the estimated source-coded parameters.

19. A method for decoding source signals as claimed in claim 17, wherein further quantization steps are used for the quantization of the estimated source-coded parameters in addition to quantization steps used for quantization of the transmitted source-coded parameters.

20. A method for decoding source signals as claimed in claim 17, wherein the quantization of the estimated source-coded parameters is carried out as a function of a current coding rate of the transmitter.

21. A method for decoding source signals as claimed in claim 16, wherein the received parameters and the reliability information are used at a receiving end, for at least one of the source-coded parameters which are possible at a transmission end, to determine a probability with which this has actually been transmitted, and the transmitted source-coded parameters are estimated taking account of the probability.

22. A method for decoding source signals as claimed in claim 16, wherein a source-coded parameter is coded in the form of a bit combination, an association bit error probability is determined for each bit, the transmitted source-coded parameter is estimated using the associated bit error probability, and the estimated source-coded parameter is quantized and passed on in the form of a corresponding bit combination.

23. A method for decoding source signals as claimed in claim 16, wherein the source-coded parameters are coded in accordance with a GSM Standard.

24. An apparatus for decoding coded source signals which are transmitted via a transmission channel, comprising:

a channel decoder which initially decodes received signals and, in the process, determines source-coded parameters from the received signals and obtains reliability information which is correlated with transmission quality;

a source decoder arranged physically separately from the channel decoder, the source-coded parameters being passed on to the source-decoder which further decodes the source-coded parameters; and

an error concealment device, which estimates transmitted source-coded parameters from the received source-coded parameters, taking account of the reliability information, wherein the error concealment device is arranged at a same location as the channel decoder, and the estimated source-coded parameters are passed on to the source decoder.

25. An apparatus for decoding coded source signals which are transmitted via a transmission channel as claimed in claim 24, further comprising:

a quantizer which quantizes the estimated source-coded parameters for passing them on to the source decoder.

26. An apparatus for decoding coded source signals which are transmitted via a transmission channel as claimed in claim 25, wherein the quantizer has at least the same quantization steps as a quantizer which quantizes the source-coded parameters, before transmission, at a transmission end.

27. An apparatus for decoding coded source signals which are transmitted via a transmission channel as claimed in claim 26, wherein the quantizer has more quantization steps than the quantizer located at the transmission end.

28. An apparatus for decoding coded source signals which are transmitted via a transmission channel as claimed in claim 25, wherein the quantizer is matched to different coding rates of the transmitter.

29. An apparatus for decoding coded source signals which are transmitted via a transmission channelas claimed in claim 24, wherein the channel decoder is arranged at a base transceiver station in a mobile radio network.

30. An apparatus for decoding coded source signals which are transmitted via a transmission channel as claimed in claim 24, wherein the quantizer is a standard GSM quantizer.

REMARKS

The present amendment makes editorial changes and corrects typographical errors in the specification, which includes the Abstract, in order to conform the specification to the requirements of United States Patent Practice. No new matter is added thereby. Attached hereto is a marked-up version of the changes made to the specification by the present amendment. The attached page is captioned "**Version With Markings To Show Changes Made**".

In addition, the present amendment cancels original claims 1-15 in favor of new claims 16-30. Claims 16-30 have been presented solely because the revisions by red-lining and underlining which would have been necessary in claims 1-15 in order to present those claims in accordance with preferred United States Patent Practice would have been too extensive, and thus would have been too burdensome. The present amendment is intended for clarification purposes only and not for substantial reasons related to patentability pursuant

to 35 USC §§101, 102, 103 or 112. Indeed, the cancellation of claims 1-15 does not constitute an intent on the part of the Applicants to surrender any of the subject matter of claims 1-15.

Early consideration on the merits is respectfully requested.

Respectfully submitted,

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VERSIONS WITH MARKINGS TO SHOW CHANGES MADE

In The Specification:

The Specification of the present application, including the Abstract, has been amended as follows:

~~Description~~

SPECIFICATION

~~Method and apparatus for decoding source signals~~

TITLE OF THE INVENTION

METHOD AND APPARATUS FOR DECODING SOURCE SIGNALS

BACKGROUND OF THE INVENTION

The present invention relates to a method for decoding source signals, which have been transmitted in coded form via a transmission channel. The present invention furthermore relates to a corresponding apparatus for decoding the source signals. The expression source signals ~~means~~ refers to, for example, voice, audio or video signals. The following text is largely based on voice signals. In this case, however, voice signals should be regarded only as an example and, in this respect, are not associated with any restriction whatsoever to the present invention.

Methods and apparatuses such as these are currently ~~normally used~~ used, typically, for voice transmission in mobile radio networks. Mobile radio networks are generally set up such that a number of receivers or receiver/transmitter stations, which are referred to as base transceiver stations (and also as BTS in the following text) are arranged as far as possible covering an area at predetermined intervals from one another, so that the reception areas of these BTS overlap. A number of these BTS are normally linked via cables to a common base station controller (also referred to as a BSC in the following text), with the distance between the BTS and the BSC generally being several kilometers. ~~These~~ In turn, these BSC are then, generally ~~in turn~~, connected to a mobile switching center (also referred to as an MSC in the following text), which is responsible, inter alia, for call management, call monitoring and for interaction with other networks~~,~~; for example, normal landline networks or the ISDN.

For transmission via a radio path, the voice signals are initially broken down in a source coder, in this case a voice coder, in time voice sections, which can be described by different speech parameters. These real-value speech parameters are then quantized. A parameter quantized in this way corresponds to an entry in a quantization table or a code book, with the parameter being represented by a specific bit combination. The speech

parameters which have been digitally coded in this way are then passed on. Other types of source signals are broken down in a similar way into source-coded parameters or else source-coded coefficients, which are then passed on.

Since interference and losses must be expected when transmitting the parameters via a mobile radio channel, ~~further,~~ redundant information is added to these coded signals in a channel coder, from which information it is possible to draw conclusions about the correctness of the received signal, at the receiver end.

The data received via the radio channel is then ~~first of all~~ initially decoded in a corresponding channel decoder, with all the data which was required only for transmission on the radio channel being separated out again. This includes, inter alia, ~~said~~ the redundant information, which contains details about the transmission quality. This channel decoder is generally located at the same place as the BTS. The speech parameters are, finally, passed on for final decoding to a source decoder (or voice decoder). This source decoder is normally a component of a TRAU (Transcoder/Rate Adapter Unit), which is arranged physically separately from the BTS, typically in the BSC or MSC. ~~This means that~~ As such, the data rate between the BTS and the BSC or the MSC is kept low, thus making it possible, inter alia, to reduce the transmission costs.

When transmitting voice data via a radio channel that is subject to severe interference, residual bit errors can still remain in the bit stream despite error correction in the course of the channel decoding process. Voice decoding based on these speech parameters that are subject to interference would generally lead to considerable distortion of the output voice signal. Additional error concealment measures are therefore required, in order to improve the subjectively perceived reproduction quality, with the information determined by the channel decoder about the respective transmission quality or the reliability of the decoded bits sensibly being used for this purpose.

In the already existing GSM Standard, channel-dependent reliability information, referred to as a BFI flag (bad frame indication), is produced in the channel decoding process for each voice frame~~,~~; that is ~~to say~~, for each received bit sequence. To do this, the channel decoder carries out a CRC (Cyclic Redundancy Check), which is formed by the auditively most important bits of the speech parameter indices~~, that is to say~~ or those bits which are most sensitive to interference. This check results in a binary decision (BFI=0, good frame; BFI=1, bad frame), which requires only one bit. This flag is passed on to the voice decoder, where it is included in the decoding process in such a way that, for a frame which is identified as being bad, the values of the last good frame are repeated directly or in slightly

modified form~~;~~, for example, by being attenuated. Muting is carried out in the event of a number of successive bad frames as a result of which, in the event of severe channel interference, so many signal components are cut off in some circumstances that the comprehensibility is considerably adversely affected.

T. Fingscheidt and P. Vary have proposed a completely new error concealment method in the article "Error Concealment by Softbit Speech Decoding", in the Proceedings of the ITG Conference Voice Communication, pages 7-10, Frankfurt am Main, September 1996. In this method, the channel-dependent reliability information determined by the channel decoder is used in order to calculate the probability with which a specific speech parameter has been transmitted, or in order to establish an estimated value for that speech parameter such that it matches the actually transmitted speech parameter as well as possible. In detail, a calculation is, in this case, carried out for each potentially transmitted bit combination to determine the probability with which this bit combination can be changed to the received bit combination. The reliability information is, in this case, determined in the form of individual bit error probabilities, with one, and only one bit error probability being associated with each individual information bit. The pair, ~~comprising~~ including the information bit itself and the associated bit error probability, is referred to as a softbit. These softbits must be made available to the error concealment device~~,~~ in order to estimate the speech parameter. In this type of error concealment, the estimation results in different real-value parameter values (intermediate values) than those contained in the transmission-end quantization table, so that the estimated parameter value cannot be coded correctly using this quantization table. For this reason, this device normally has, until now ~~normally,~~ been connected directly to the voice decoder, since it can process the estimated real-value parameter values directly~~,~~ without any additional losses.

However, this leads to difficulties at the base station end. For the reasons mentioned above, the channel decoder, which obtains the reliability information from the transmitted signals, is located at the same point as the receiver station (BTS) while the voice decoder is located at the TRAU, with the transmission rate being limited on the connection between them. With the current GSM Standard, the limit is 16 or 8 kbit/s (full rate or half rate), which corresponds to 16 or 8-sub-multiplexing, respectively. However, if a softbit is represented, for example, by 4 bits in an error concealment process based on the last described method, this would result in a data stream at a total of 12.2 x 4 = 48.8 kbps when using the normal 12.2 kbps voice coders and decoders in accordance with the GSM Standard. Transmission via a data line which is limited to 16 kbps is thus impossible.

~~The~~ An object of the present invention is, therefore, to provide an alternative to this prior art.

~~This object is achieved by a method having the features of patent claim 1, and by an apparatus having the features of claim 9.~~

SUMMARY OF THE INVENTION

~~The critical~~ A key feature of the ~~new~~ inventive method and ~~of the new~~ apparatus is the decoupling~~, according to the invention,~~ of the error concealment from the actual source decoding. By carrying out the error concealment at the same location as the channel decoding, or the receiver, there is no longer any need to pass on additional channel status information to the source decoder in addition to the source-coded parameters, so that the data stream can be reduced considerably. It is thus possible, even with the existing systems and with the previous standards, to carry out an improved error concealment method, which uses the channel-dependent reliability information more effectively.

The source-coded parameters which are estimated in the error concealment method can, in principle, be passed on to the source decoder in any desired manner. However, preferably, these parameters are quantized in the same way as the transmitted source-coded parameters and are passed on in the known digital form to the source decoder~~,~~ in order to minimize the transmitted data streams. If the quantization is chosen in an appropriate suitable manner, it has been found that this renewed quantization, referred to as requantization in the following text, does not result in any additional losses or interference whatsoever.

In particular, it has been found that it is advantageous if at least the same quantization steps as those used for quantization of the originally transmitted source-coded parameters are used for quantization of the estimated source-coded parameters. In the event of error-free transmission, the transmitted source-coded parameters are ~~thus~~ automatically reproduced exactly. If this were not the case, then errors could occur due to the requantization in the event of transmission without interference.

Various investigations and simulations have shown that, when using quantizers with a sufficiently high quantization rate R, it is sufficient to use exactly the same quantizers as those at the transmission end for requantization of the estimated parameters. If the parameters have a Gaussian distribution, this is generally possible for R > 1 bit/sample value for transmitter-end quantizers. The Gaussian distribution criterion is, in this case ~~generally~~, to an approximation, quite well satisfied by most of the transmitted source-coded parameters.

The code book which is used at the transmission end ~~can~~ thus can be used as the quantization code book for the estimated parameter.

If a quantizer with a rate of $R \leq 1$ bit/sample value is used at the transmission end, any influence from the quantization of the estimated parameters can be ameliorated by using a modified quantizer at a higher rate at this point. ~~This means that~~ As such, a quantizer is chosen which uses further quantization steps in addition to the quantization steps which are available at the transmission end. For example, if a quantizer with a rate of only $R = 1$ bit/sample value is used at the transmission end, it is worthwhile using a quantizer with a rate of $R = 2$ bits/sample value for quantization of the estimated parameters. This allows additional interference due to requantization to be avoided very well at the receiver end, even in cases such as this, in a simple manner.

In the course of the further development of the GSM mobile radio standard, a new standard for coded voice transmission has now been developed. This relates to coders and decoders (codecs) which allow the overall available data rate to be split in a manner matched to the channel state and to the system load (adaptive multirate codecs; AMR codecs). In this case, the channel mode is defined on the one hand (full rate 22.8 kbps or half rate 11.4 kbps) while, on the other hand, the source and channel coding are matched to the data rate that is then available. ~~This means that~~ As such, the coding rate is varied during a transmission as a function of the transmission quality of the channel and the number of users who are using this channel at the same time. The number of quantization steps for the coding process is also changed in a corresponding manner. It is, thus, advantageous for the quantization of the source-coded parameters, which are estimated using the error concealment method, to be carried out as a function of the present coding rate of the transmitter~~,~~ as well. This is preferably done using a quantizer which is matched to the various coding rates of the transmitter.

~~The invention will be explained in more detail in the following text, using an exemplary embodiment and with reference to the attached drawings, in which:~~

Additional features and advantages of the present invention are described in, and will be apparent from, the following Detailed Description of the Invention and the Figures.

BRIEF DESCRIPTION OF THE FIGURES

Figure 1 shows a schematic illustration of the position and connections for the receivers (BTS), the BSC and the mobile switching station (MSC)~~,~~.

Figure 2 shows a schematic illustration of the signal path through the individual coders and decoders.

DETAILED DESCRIPTION OF THE INVENTION

As is shown in simplified form in Figure 1, a conventional modern base station operating in accordance with the GSM mobile radio standard ~~comprises~~ includes a BSC 11, to which a number of BTS 6 are connected via data lines 14. The BSC 11 is, in turn, connected via a data line 15 to an MSC 13. In general, one MSC 13 serves a number of BSC 11, although this is not illustrated in Figure 1, for space reasons.

The BTS 6 are located spaced apart from one another such that their reception areas 17 just overlap, so that the supply from the BTS 6 covers the area as well as possible. If a mobile telephone 1 is now located in the reception area of one BTS 6, then it can communicate with this BTS 6 via a radio path 17. When an active mobile telephone 1 leaves the reception area 17 of one BTS 6 and enters the reception area 17 of another BTS 6 associated with the same base station, ~~then~~ this is identified automatically by the associated BTS 11, and a handover takes place from the one BTS 6 to the next. During a handover from the area of one base station to another base station, the handover process is carried out with the aid of the MSC 13~~; this means that~~. As such, the call is automatically handed over to the new base station, with the other BSC 11.

The details of the profile of a voice signal from a mobile telephone 1 to the BSC 11, in particular the different coding and decoding stages, can be seen from the simplified schematic illustration in Figure 2. This shows only the signal interchange in one direction. Normally, the coders and decoders are each codecs, that is to say combined coding/decoding devices, and the transmitting and receiving units are combined transmitting/receiving units (transceivers).

As illustrated in Figure 2, the voice signal is first of all passed through a voice coder 2, in which the speech is broken down into individual speech parameters. Each sound is, in this case, represented by a specific number of speech parameters. Typical speech parameters in one representation of the voice signals are, for example, the "LPC coefficient," the "LTP index," the "LTP gain," the "codebook indices" and the "codebook gain," and in another representation are the "LSP set," the "pitch delay," the "pitch gain," the "algebraic code" and the "codebook gain."

These speech parameters are then passed through a quantizer 3, where they are converted to a bit combination~~,~~; that is to say, a real-value speech parameter v is represented by the bit combination X after quantization. Depending on the transmission rate, a different

number of bits are available for coding for the individual parameters. In the AMR Standard, eight different transmission modes, with data rates of between 12.2 kbps and 4.75 kbps can be used for voice transmission.

The speech parameters v are transmitted from the quantizer 3, in the digital representation X, to a channel coder 4, which adds the channel information, required for transmission, to the data. This includes redundant data, which allows the receiver to check the correctness of the received data and, if necessary, to correct transmission errors. The radio signals are then sent from a transmission unit 5 or a transceiver via a radio channel 16 to a receiving element 7, for example an antenna with a demodulator and/or equalizer, of the BTS 6.

In the BTS 6, the received signals S are first of all passed to a channel decoder 8, which initially decodes the received signals S. During the initial decoding process, reliability information which is correlated with the transmission quality is obtained from the signals S. This may ~~comprise~~ include, for example, the results of parity checks.

This additionally obtained information about the channel state is transmitted together with the received speech parameters determined in the channel decoder 8 to an error concealment device 9. This is indicated in Figure 2 by two arrows between the channel decoder 8 and the error concealment device 9. The speech parameter received via the radio channel is still available in digitally coded form at the output of the channel decoder 8, that is to say it is represented by a bit combination X', which matches the transmitted bit combination X, provided the transmission has taken place without errors.

In the present exemplary embodiment, the error concealment device 9 operates in accordance with the error concealment method cited in the prior art by T. Fingscheidt and P. Vary. In this method, the reliability information is initially used to determine, for each individual bit in the combination X, the probability that it has been transmitted without any errors. This probability is dependent solely on the state of the channel. In addition, the error concealment method can also make use of information which is dependent on the original source of the received signal. This may be, for example, the probabilities with which a specific parameter, and hence a specific bit combination, can occur at the transmission end. These probabilities of occurrence can be determined in advance ~~by means of~~ via a representative speech database, and can be stored in tables. Furthermore, it also would ~~also~~ be possible to take account of the probability of two specific parameters or bit combinations following one another directly, in order ~~thus also~~ to take account of correlations between successive received voice frames.

~~Said~~ The individual probabilities, which are either channel-dependent or source-dependent, are then used either to estimate a parameter $\hat{v}$, which most probably corresponds to the originally transmitted speech parameter v (maximum a-posteriori estimation) or an estimate is made, in which the mean square error of the possible errors between the estimated parameter $\hat{v}$ and the transmitted parameter v is a minimum (mean-square estimation).

The real-value parameter $\hat{v}$ estimated by the error concealment device 9 is then once again passed (before being transmitted to the voice decoder 12 arranged at the BSC 11) through a quantizer 10 which operates in the same way as the quantizer 3 at the transmission end and converts the real-value parameter $\hat{v}$ to a digital bit combination $\hat{X}$. Instead of the bit combination X' determined by the channel decoder for the received parameter, the voice decoder 12, in consequence, now receives a bit combination $\hat{X}$, which is coded in the same way and represents the speech parameter $\hat{v}$ which is estimated by the error concealment and which most probably matches the transmitted speech parameter v, which differs from it by the smallest error.

The quantizer 10 in the BTS in the present exemplary embodiment is precisely the same model as the quantizer 3 in the mobile telephone 1. These are normal codecs~~,~~; for example, AMR or FR (full rate) codecs to the GSM Standard. The codebooks available to the quantizer 10 are also the same as those available to the quantizer 3.

It should be mentioned once again that the present invention is not restricted to the specific exemplary embodiment described. Thus, in principle, it is also possible to use it in systems which do not operate in accordance with the GSM Standard, but, for example, in accordance with more recent Standards that are currently still being developed, such as the UMTS Standard. In the same way, the source decoder 12 need not necessarily be located in the BSC 11, but can be arranged as an autonomous unit, on its own or combined with other functional units, for example in the form of what is referred to as the TRAU, upstream or downstream of the BSC. In particular, instead of the method according to Fingscheidt and Vary, it is also possible to use a different error concealment method, which uses the reliability information to estimate the transmitted source-coded parameter, that is to say which corrects the received parameter using the reliability information, such that it corresponds to the supposedly correct transmitted parameter. Furthermore, as already mentioned above, the present invention is not restricted to voice signals, but ~~can~~ also can be used for any other desired source signals.

The method and the apparatus according to the present invention result in improved transmission quality even in very poor radio channels, since the channel-dependent reliability information received by the channel decoder and the source-dependent information are used in a considerably better way to eliminate errors that occur. Owing to the specific physical arrangement of the error concealment device and the downstream quantizers, this can be done without increasing the data rate between the channel decoder and the source decoder.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

~~Abstract~~

~~Method and apparatus for decoding source signals~~

ABSTRACT OF THE DISCLOSURE

A method and an apparatus ~~are described~~ for decoding coded source signals which are transmitted via a transmission channel. In this case, the received signals ~~(S)~~ are ~~first of all~~ initially decoded in a channel decoder ~~(8)~~ and, in the process, source-coded parameters are determined from the received signals ~~(S)~~ and are passed on to a source decoder~~(12)~~, which is physically separated from the channel decoder~~(8)~~, where they are further-decoded. Reliability information, which is correlated with the transmission quality, is obtained from the received signals ~~(S)~~ during the initial decoding. The transmitted source-coded parameters are estimated from the received source-coded parameters ~~by means of~~ via this reliability information using an error concealment method. The error concealment is carried out at the same location as the channel decoder~~(8)~~, and the estimated source-coded parameters are passed on to the source decoder~~(12)~~.

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Description

Method and apparatus for decoding source signals

The invention relates to a method for decoding source signals, which have been transmitted in coded form via a transmission channel. The invention furthermore relates to a corresponding apparatus for decoding the source signals. The expression source signals means, for example, voice, audio or video signals. The following text is largely based on voice signals. In this case, however, voice signals should be regarded only as an example and, in this respect, are not associated with any restriction whatsoever to the invention.

Methods and apparatuses such as these are currently normally used for voice transmission in mobile radio networks. Mobile radio networks are generally set up such that a number of receivers or receiver/transmitter stations, which are referred to as base transceiver stations (and also as BTS in the following text) are arranged as far as possible covering an area at predetermined intervals from one another, so that the reception areas of these BTS overlap. A number of these BTS are normally linked via cables to a common base station controller (also referred to as a BSC in the following text), with the distance between the BTS and the BSC generally being several kilometers. These BSC are then generally in turn connected to a mobile switching center (also referred to as an MSC in the following text), which is responsible, inter alia, for call management, call monitoring and for interaction with other networks, for example normal landline networks or the ISDN.

For transmission via a radio path, the voice signals are initially broken down in a source coder, in this

case a voice coder, in time voice sections, which can be described by different speech

parameters. These real-value speech parameters are then quantized. A parameter quantized in this way corresponds to an entry in a quantization table or a code book, with the parameter being represented by a specific bit combination. The speech parameters which have been digitally coded in this way are then passed on. Other types of source signals are broken down in a similar way into source-coded parameters or else source-coded coefficients, which are then passed on.

Since interference and losses must be expected when transmitting the parameters via a mobile radio channel, further, redundant information is added to these coded signals in a channel coder, from which information it is possible to draw conclusions about the correctness of the received signal, at the receiver end.

The data received via the radio channel is then first of all initially decoded in a corresponding channel decoder, with all the data which was required only for transmission on the radio channel being separated out again. This includes, inter alia, said redundant information, which contains details about the transmission quality. This channel decoder is generally located at the same place as the BTS. The speech parameters are, finally, passed on for final decoding to a source decoder (or voice decoder). This source decoder is normally a component of a TRAU (Transcoder/Rate Adapter Unit), which is arranged physically separately from the BTS, typically in the BSC or MSC. This means that the data rate between the BTS and the BSC or the MSC is kept low, thus making it possible, inter alia, to reduce the transmission costs.

When transmitting voice data via a radio channel that is subject to severe interference, residual bit errors can still remain in the bit stream despite

error correction in the course of the channel decoding process. Voice decoding based on these speech parameters that are subject to interference would generally lead to considerable distortion of the output voice signal. Additional error concealment measures are therefore required, in order to improve the subjectively perceived reproduction quality, with the information determined by the channel decoder about the respective transmission quality or the reliability of the decoded bits sensibly being used for this purpose.

In the already existing GSM Standard, channel-dependent reliability information, referred to as a BFI flag (bad frame indication), is produced in the channel decoding process for each voice frame, that is to say for each received bit sequence. To do this, the channel decoder carries out a CRC (Cyclic Redundancy Check), which is formed by the auditively most important bits of the speech parameter indices, that is to say those bits which are most sensitive to interference. This check results in a binary decision (BFI=0, good frame; BFI=1, bad frame), which requires only one bit. This flag is passed on to the voice decoder, where it is included in the decoding process in such a way that, for a frame which is identified as being bad, the values of the last good frame are repeated directly or in slightly modified form, for example by being attenuated. Muting is carried out in the event of a number of successive bad frames as a result of which, in the event of severe channel interference, so many signal components are cut off in some circumstances that the comprehensibility is considerably adversely affected.

T. Fingscheidt and P. Vary have proposed a completely new error concealment method in the article "Error Concealment by Softbit Speech Decoding", in the Proceedings of the ITG Conference Voice Communication, pages 7-10, Frankfurt am Main, September 1996. In this method, the channel-dependent reliability information

determined by the channel decoder is used in order to calculate the

probability with which a specific speech parameter has been transmitted, or in order to establish an estimated value for that speech parameter such that it matches the actually transmitted speech parameter as well as possible. In detail, a calculation is in this case carried out for each potentially transmitted bit combination to determine the probability with which this bit combination can be changed to the received bit combination. The reliability information is in this case determined in the form of individual bit error probabilities, with one, and only one bit error probability being associated with each individual information bit. The pair, comprising the information bit itself and the associated bit error probability, is referred to as a softbit. These softbits must be made available to the error concealment device, in order to estimate the speech parameter. In this type of error concealment, the estimation results in different real-value parameter values (intermediate values) than those contained in the transmission-end quantization table, so that the estimated parameter value cannot be coded correctly using this quantization table. For this reason, this device has until now normally been connected directly to the voice decoder, since it can process the estimated real-value parameter values directly, without any additional losses.

However, this leads to difficulties at the base station end. For the reasons mentioned above, the channel decoder, which obtains the reliability information from the transmitted signals, is located at the same point as the receiver station (BTS) while the voice decoder is located at the TRAU, with the transmission rate being limited on the connection between them. With the current GSM Standard, the limit is 16 or 8 kbit/s (full rate or half rate), which corresponds to 16 or 8-sub-multiplexing, respectively. However, if a softbit is represented, for example, by 4 bits in an error concealment process

based on the last described method, this would result in a data stream at a total of 12.2 x 4 = 48.8 kbps when using the normal 12.2 kbps voice coders and decoders in accordance with the GSM Standard. Transmission via a data line which is limited to 16 kbps is thus impossible.

The object of the present invention is to provide an alternative to this prior art.

This object is achieved by a method having the features of patent claim 1, and by an apparatus having the features of claim 9.

The critical feature of the new method and of the new apparatus is the decoupling, according to the invention, of the error concealment from the actual source decoding. By carrying out the error concealment at the same location as the channel decoding, or the receiver, there is no longer any need to pass on additional channel status information to the source decoder in addition to the source-coded parameters, so that the data stream can be reduced considerably. It is thus possible, even with the existing systems and with the previous standards, to carry out an improved error concealment method, which uses the channel-dependent reliability information more effectively.

The source-coded parameters which are estimated in the error concealment method can, in principle, be passed on to the source decoder in any desired manner. However, preferably, these parameters are quantized in the same way as the transmitted source-coded parameters and are passed on in the known digital form to the source decoder, in order to minimize the transmitted data streams. If the quantization is chosen in an appropriate suitable manner, it has been found that this renewed quantization, referred to as

requantization in the following text, does not result in any additional losses or interference whatsoever.

In particular, it has been found that it is advantageous if at least the same quantization steps as those used for quantization of the originally transmitted source-coded parameters are used for quantization of the estimated source-coded parameters. In the event of error-free transmission, the transmitted source-coded parameters are thus automatically reproduced exactly. If this were not the case, then errors could occur due to the requantization in the event of transmission without interference.

Various investigations and simulations have shown that, when using quantizers with a sufficiently high quantization rate R, it is sufficient to use exactly the same quantizers as those at the transmission end for requantization of the estimated parameters. If the parameters have a Gaussian distribution, this is generally possible for $R > 1$ bit/sample value for transmitter-end quantizers. The Gaussian distribution criterion is in this case generally, to an approximation, quite well satisfied by most of the transmitted source-coded parameters. The code book which is used at the transmission end can thus be used as the quantization code book for the estimated parameter.

If a quantizer with a rate of $R \leq 1$ bit/sample value is used at the transmission end, any influence from the quantization of the estimated parameters can be ameliorated by using a modified quantizer at a higher rate at this point. This means that a quantizer is chosen which uses further quantization steps in addition to the quantization steps which are available at the transmission end. For example, if a quantizer with a rate of only $R = 1$ bit/sample value is used at the transmission end, it is worthwhile using a

quantizer with a rate of R = 2 bits/sample value for quantization

of the estimated parameters. This allows additional interference due to requantization to be avoided very well at the receiver end, even in cases such as this, in a simple manner.

In the course of the further development of the GSM mobile radio standard, a new standard for coded voice transmission has now been developed. This relates to coders and decoders (codecs) which allow the overall available data rate to be split in a manner matched to the channel state and to the system load (adaptive multirate codecs; AMR codecs). In this case, the channel mode is defined on the one hand (full rate 22.8 kbps or half rate 11.4 kbps) while, on the other hand, the source and channel coding are matched to the data rate that is then available. This means that the coding rate is varied during a transmission as a function of the transmission quality of the channel and the number of users who are using this channel at the same time. The number of quantization steps for the coding process is also changed in a corresponding manner. It is thus advantageous for the quantization of the source-coded parameters, which are estimated using the error concealment method, to be carried out as a function of the present coding rate of the transmitter, as well. This is preferably done using a quantizer which is matched to the various coding rates of the transmitter.

The invention will be explained in more detail in the following text, using an exemplary embodiment and with reference to the attached drawings, in which:

Figure 1 shows a schematic illustration of the position and connections for the receivers (BTS), the BSC and the mobile switching station (MSC),

Figure 2 shows a schematic illustration of the signal path through the individual coders and decoders.

As is shown in simplified form in Figure 1, a conventional modern base station operating in accordance with the GSM mobile radio standard comprises a BSC 11, to which a number of BTS 6 are connected via data lines 14. The BSC 11 is in turn connected via a data line 15 to an MSC 13. In general, one MSC 13 serves a number of BSC 11, although this is not illustrated in Figure 1, for space reasons.

The BTS 6 are located spaced apart from one another such that their reception areas 17 just overlap, so that the supply from the BTS 6 covers the area as well as possible. If a mobile telephone 1 is now located in the reception area of one BTS 6, then it can communicate with this BTS 6 via a radio path 17. When an active mobile telephone 1 leaves the reception area 17 of one BTS 6 and enters the reception area 17 of another BTS 6 associated with the same base station, then this is identified automatically by the associated BTS 11, and a handover takes place from the one BTS 6 to the next. During a handover from the area of one base station to another base station, the handover process is carried out with the aid of the MSC 13; this means that the call is automatically handed over to the new base station, with the other BSC 11.

The details of the profile of a voice signal from a mobile telephone 1 to the BSC 11, in particular the different coding and decoding stages, can be seen from the simplified schematic illustration in Figure 2. This shows only the signal interchange in one direction. Normally, the coders and decoders are each codecs, that is to say combined coding/decoding devices, and the transmitting and receiving units are combined transmitting/receiving units (transceivers).

As illustrated in Figure 2, the voice signal is first of all passed through a voice coder 2, in which the speech

is broken down into individual speech parameters. Each sound is in this case represented by a specific number of speech parameters. Typical speech parameters in one representation of the voice signals are, for example, the "LPC coefficient", the "LTP index", the "LTP gain" and the "codebook indices" and the "codebook gain", and in another representation are the "LSP set", the "pitch delay", the "pitch gain", the "algebraic code" and the "codebook gain".

These speech parameters are then passed through a quantizer 3, where they are converted to a bit combination, that is to say a real-value speech parameter v is represented by the bit combination X after quantization. Depending on the transmission rate, a different number of bits are available for coding for the individual parameters. In the AMR Standard, eight different transmission modes, with data rates of between 12.2 kbps and 4.75 kbps can be used for voice transmission.

The speech parameters v are transmitted from the quantizer 3, in the digital representation X, to a channel coder 4, which adds the channel information, required for transmission, to the data. This includes redundant data, which allows the receiver to check the correctness of the received data and, if necessary, to correct transmission errors. The radio signals are then sent from a transmission unit 5 or a transceiver via a radio channel 16 to a receiving element 7, for example an antenna with a demodulator and/or equalizer, of the BTS 6.

In the BTS 6, the received signals S are first of all passed to a channel decoder 8, which initially decodes the received signals S. During the initial decoding process, reliability information which is correlated with the transmission quality is obtained from the

signals S. This may comprise, for example, the results of parity checks.

This additionally obtained information about the channel state is transmitted together with the received speech parameters determined in the channel decoder 8 to an error concealment device 9. This is indicated in Figure 2 by two arrows between the channel decoder 8 and the error concealment device 9. The speech parameter received via the radio channel is still available in digitally coded form at the output of the channel decoder 8, that is to say it is represented by a bit combination X', which matches the transmitted bit combination X, provided the transmission has taken place without errors.

In the present exemplary embodiment, the error concealment device 9 operates in accordance with the error concealment method cited in the prior art by T. Fingscheidt and P. Vary. In this method, the reliability information is initially used to determine, for each individual bit in the combination X, the probability that it has been transmitted without any errors. This probability is dependent solely on the state of the channel. In addition, the error concealment method can also make use of information which is dependent on the original source of the received signal. This may be, for example, the probabilities with which a specific parameter, and hence a specific bit combination, can occur at the transmission end. These probabilities of occurrence can be determined in advance by means of a representative speech database, and can be stored in tables. Furthermore, it would also be possible to take account of the probability of two specific parameters or bit combinations following one another directly, in order thus also to take account of correlations between successive received voice frames.

Said individual probabilities, which are either channel-dependent or source-dependent, are then used either to estimate a parameter $\hat{v}$, which most probably

corresponds to the originally transmitted speech parameter v (maximum a-posteriori estimation) or an estimate is made, in which the mean square error of the possible errors between the estimated parameter $\hat{v}$ and the transmitted parameter v is a minimum (mean-square estimation).

The real-value parameter $\hat{v}$ estimated by the error concealment device 9 is then once again passed (before being transmitted to the voice decoder 12 arranged at the BSC 11) through a quantizer 10 which operates in the same way as the quantizer 3 at the transmission end and converts the real-value parameter $\hat{v}$ to a digital bit combination $\hat{X}$. Instead of the bit combination X' determined by the channel decoder for the received parameter, the voice decoder 12 in consequence now receives a bit combination $\hat{X}$, which is coded in the same way and represents the speech parameter $\hat{v}$ which is estimated by the error concealment and which most probably matches the transmitted speech parameter v, which differs from it by the smallest error.

The quantizer 10 in the BTS in the present exemplary embodiment is precisely the same model as the quantizer 3 in the mobile telephone 1. These are normal codecs, for example AMR or FR (full rate) codecs to the GSM Standard. The codebooks available to the quantizer 10 are also the same as those available to the quantizer 3.

It should be mentioned once again that the invention is not restricted to the specific exemplary embodiment described. Thus, in principle, it is also possible to use it in systems which do not operate in accordance with the GSM Standard, but, for example, in accordance with more recent Standards that are currently still being developed, such as the UMTS Standard. In the same way, the source decoder 12 need not necessarily be

located in the BSC 11, but can be arranged as an autonomous unit,

on its own or combined with other functional units, for example in the form of what is referred to as the TRAU, upstream or downstream of the BSC. In particular, instead of the method according to Fingscheidt and Vary, it is also possible to use a different error concealment method, which uses the reliability information to estimate the transmitted source-coded parameter, that is to say which corrects the received parameter using the reliability information, such that it corresponds to the supposedly correct transmitted parameter. Furthermore, as already mentioned above, the invention is not restricted to voice signals, but can also be used for any other desired source signals.

The method and the apparatus according to the invention result in improved transmission quality even in very poor radio channels, since the channel-dependent reliability information received by the channel decoder and the source-dependent information are used in a considerably better way to eliminate errors that occur. Owing to the specific physical arrangement of the error concealment device and the downstream quantizers, this can be done without increasing the data rate between the channel decoder and the source decoder.

Patent Claims

1. A method for decoding source signals, in which the received signals (S) are first of all initially decoded in a channel decoder (8) and, in the process, source-coded parameters are determined from the received signals (S) and are passed on to a source decoder (12), which is physically separated from the channel decoder (8), where they are further-decoded,
with reliability information, which is correlated with the transmission quality, being obtained from the received signals (S) during the initial decoding,
and with the transmitted source-coded parameters being estimated from the received source-coded parameters by means of this reliability information using an error concealment method,
characterized in that the error concealment is carried out at the same location as the channel decoder (8), and the estimated source-coded parameters are passed on to the source decoder (12).

2. The method as claimed in claim 1, characterized in that the estimated source-coded parameters are quantized for passing on to the source decoder (12).

3. The method as claimed in claim 1 or 2, characterized in that at least the same quantization steps as those used for the quantization of the transmitted source-coded parameters are used for quantization of the estimated source-coded parameters.

4. The method as claimed in one of claims 1 to 3, characterized in that further quantization steps are used for quantization of the estimated source-coded parameters, in addition to the

quantization steps which are used for quantization of the transmitted source-coded parameters.

5. The method as claimed in one of claims 1 to 4, characterized in that the quantization of the estimated source-coded parameters is carried out as a function of the current coding rate of the transmitter (1).

6. The method as claimed in one of claims 1 to 5, characterized in that the received parameter and the reliability information are used at the receiving end, for at least one of the source-coded parameters which are possible at the transmission end, to determine the probability with which this has actually been transmitted, and the transmitted source-coded parameters are then estimated taking account of these probabilities.

7. The method as claimed in one of claims 1 to 6, characterized in that a source-coded parameter is coded in the form of a bit combination, and an associated bit error probability is determined for each bit, using which the transmitted source-coded parameter is estimated, and the estimated source-coded parameter is then quantized and is passed on in the form of a corresponding bit combination.

8. The method as claimed in one of claims 1 to 7, characterized in that the source-coded parameters are coded in accordance with the GSM Standard.

9. An apparatus for decoding coded source signals which are transmitted via a transmission channel,
having a channel decoder (8) which initially decodes the received signals (S) and in the process determines source-coded parameters from the received signals (S) and obtains reliability information, which is correlated with the transmission quality,

and having a source decoder (12) which is arranged physically separately from the channel decoder (8), and to which the source-coded parameters are passed on and which further-decodes the source-coded parameters,
and having an error concealment device (9), which estimates the transmitted source-coded parameters from the received source-coded parameters, taking account of the reliability information,
characterized in that the error concealment device (9) is arranged at the same location as the channel decoder (8), and the estimated source-coded parameters are passed on to the source decoder (12).

10. The apparatus as claimed in claim 9, characterized by a quantizer (10), which quantizes the estimated source-coded parameters for passing them on to the source decoder (12).

11. The apparatus as claimed in claim 9 or 10, characterized in that the quantizer (10) has at least the same quantization steps as a quantizer (3) which quantizes the source-coded parameters, before transmission, at the transmission end.

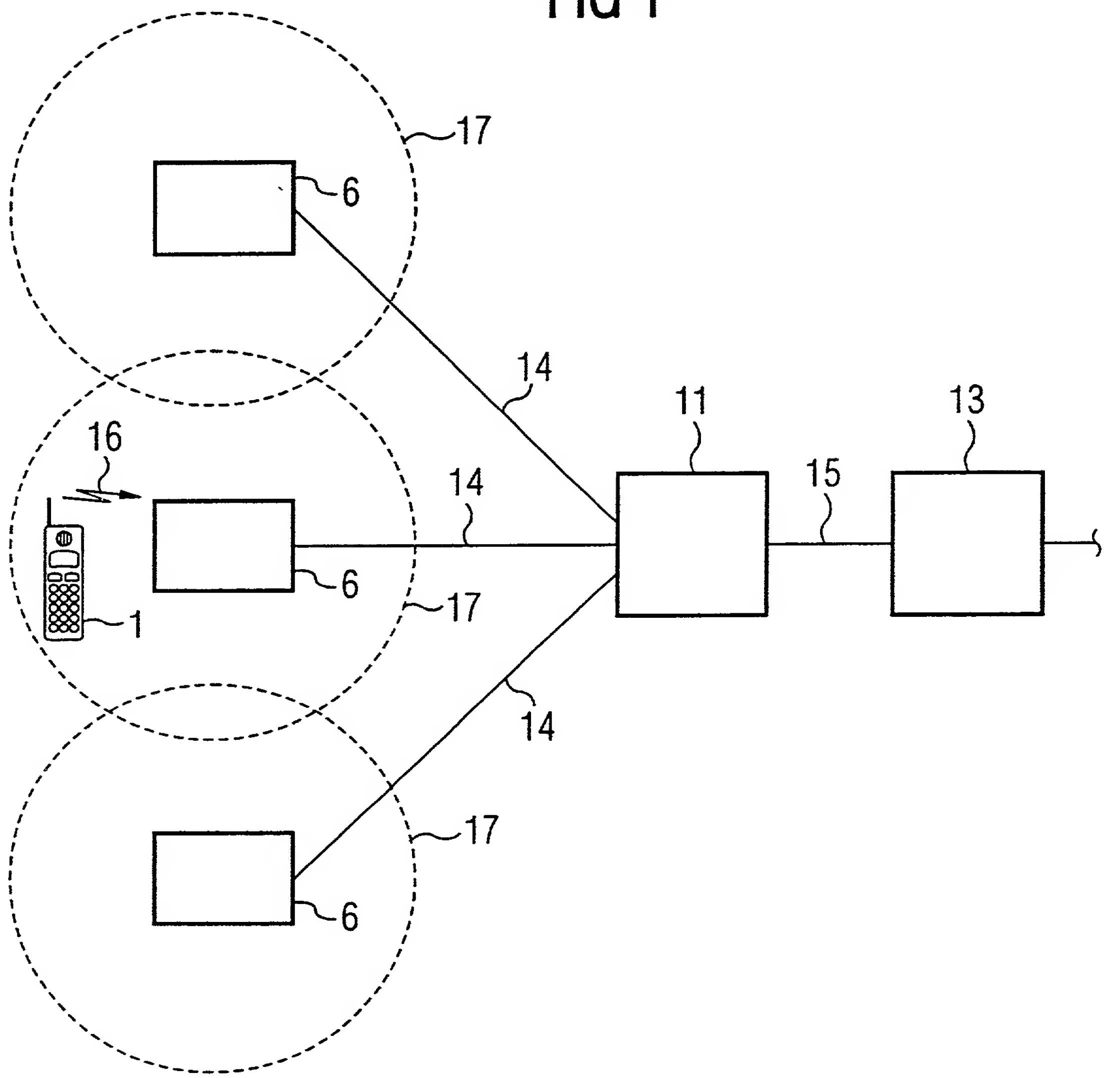
12. The apparatus as claimed in one of claims 9 to 11, characterized in that the quantizer (10) has more quantization steps than the quantizer (3) located at the transmission end.

13. The apparatus as claimed in one of claims 9 to 12, characterized in that the quantizer (10) is matched to different coding rates of the transmitter (1).

14. The apparatus as claimed in one of claims 8 to 13, characterized in that the channel decoder (8) is arranged at a base transceiver station (BTS) in a mobile radio network.

15. The apparatus as claimed in one of claims 9 to 14, characterized in that the quantizer (10) is a standard GSM quantizer.

FIG 1



10/031292

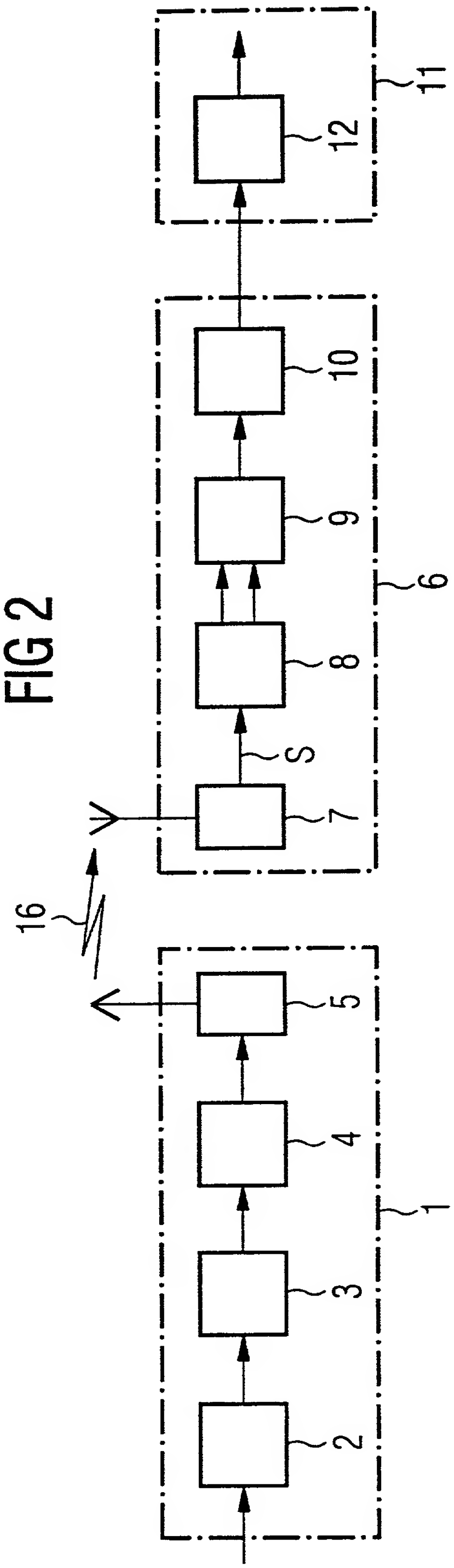


FIG 2

Declaration and Power of Attorney For Patent Application
Erklärung Für Patentanmeldungen Mit Vollmacht
German Language Declaration

Als nachstehend benannter Erfinder erkläre ich hiermit an Eides Statt:

dass mein Wohnsitz, meine Postanschrift, und meine Staatsangehörigkeit den im Nachstehenden nach meinem Namen aufgeführten Angaben entsprechen,

dass ich, nach bestem Wissen der ursprüngliche, erste und alleinige Erfinder (falls nachstehend nur ein Name angegeben ist) oder ein ursprünglicher, erster und Miterfinder (falls nachstehend mehrere Namen aufgeführt sind) des Gegenstandes bin, für den dieser Antrag gestellt wird und für den ein Patent beantragt wird für die Erfindung mit dem Titel:

Verfahren und Vorrichtung zur Decodierung von Quellensignalen

deren Beschreibung

(zutreffendes ankreuzen)
☐ hier beigefügt ist.
☒ am 14.07.2000 als
PCT internationale Anmeldung
PCT Anmeldungsnummer PCT/DE00/02303
eingereicht wurde und am _______
abgeändert wurde (falls tatsächlich abgeändert).

Ich bestätige hiermit, dass ich den Inhalt der obigen Patentanmeldung einschliesslich der Ansprüche durchgesehen und verstanden habe, die eventuell durch einen Zusatzantrag wie oben erwähnt abgeändert wurde.

Ich erkenne meine Pflicht zur Offenbarung irgendwelcher Informationen, die für die Prüfung der vorliegenden Anmeldung in Einklang mit Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) von Wichtigkeit sind, an.

Ich beanspruche hiermit ausländische Prioritätsvorteile gemäss Abschnitt 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 119 aller unten angegebenen Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde, und habe auch alle Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde nachstehend gekennzeichnet, die ein Anmeldedatum haben, das vor dem Anmeldedatum der Anmeldung liegt, für die Priorität beansprucht wird.

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

Method and device for decoding source signals

the specification of which

(check one)
☐ is attached hereto.
☒ was filed on 14.07.2000 as
PCT international application
PCT Application No. PCT/DE00/02303
and was amended on __________
(if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

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Patent and Trademark Office-U.S. DEPARTMENT OF COMMERCE
IDNR: 2590 / V: 99-1.00 / B:Val

German Language Declaration

Prior foreign appplications
Priorität beansprucht

Priority Claimed

19932943.5 (Number) (Nummer)	DE (Country) (Land)	14.07.1999 (Day Month Year Filed) (Tag Monat Jahr eingereicht)	☒ Yes Ja	☐ No Nein
(Number) (Nummer)	(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein
(Number) (Nummer)	(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein

Ich beanspruche hiermit gemäss Absatz 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 120, den Vorzug aller unten aufgeführten Anmeldungen und falls der Gegenstand aus jedem Anspruch dieser Anmeldung nicht in einer früheren amerikanischen Patentanmeldung laut dem ersten Paragraphen des Absatzes 35 der Zivilprozeßordnung der Vereinigten Staaten, Paragraph 122 offenbart ist, erkenne ich gemäss Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) meine Pflicht zur Offenbarung von Informationen an, die zwischen dem Anmeldedatum der früheren Anmeldung und dem nationalen oder PCT internationalen Anmeldedatum dieser Anmeldung bekannt geworden sind.

I hereby claim the benefit under Title 35. United States Code. §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §122, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occured between the filing date of the prior application and the national or PCT international filing date of this application.

PCT/DE00/02303 (Application Serial No.) (Anmeldeseriennummer)	14.07.2000 (Filing Date D, M, Y) (Anmeldedatum T, M, J)	anhängig (Status) (patentiert, anhängig, aufgegeben)	pending (Status) (patented, pending, abandoned)
(Application Serial No.) (Anmeldeseriennummer)	(Filing Date D,M,Y) (Anmeldedatum T, M; J)	(Status) (patentiert, anhängig, aufgeben)	(Status) (patented, pending, abandoned)

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I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

German Language Declaration

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2

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82008 UNTERHACHING	82008 UNTERHACHING

(Bitte entsprechende Informationen und Unterschriften im Falle von dritten und weiteren Miterfindern angeben).

(Supply similar information and signature for third and subsequent joint inventors).

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